In the Claims:

1. (Currently Amended) A method for suppressing background noise from a speech signal, said method comprising:

obtaining an input speech signal;

performing linear predictive coding (LPC) analysis on said input speech signal to obtain a z-domain representation of said input speech signal;

computing a spectrum tilt and a noise-to-signal ratio (NSR) of said z-domain representation of said input speech signal;

obtaining a spectrum tilt of a background noise model;

applying a gain to reduce energy of said input speech signal when said NSR is high;

reducing ~~the~~ a spectral valley energy of said input speech signal when said spectrum tilt of said input speech signal is close or equivalent to said spectrum tilt of said background noise model; and

applying an inverse filter to said input speech signal when said spectrum tilt of said input speech signal is not close ~~equivalent~~ to said spectrum tilt of said background noise model, wherein said inverse filter is an inverse of said z-domain representation of said background noise model.

2. (Previously Presented) The method of claim 1, wherein said input speech signal comprises a plurality of sub-frames processed in sequence.

3. (Currently Amended) The method of claim 1, wherein said gain is adaptive~~ly~~ based on characteristics of said input speech.

4. (Previously Presented) The method of claim 1, wherein said background noise model is a first order model.

5. (Currently Amended) A computer program product comprising:

a computer usable medium having computer readable program code embodied therein for suppressing background noise from a speech signal; said computer readable program code configured to cause a computer to:

obtain an input speech signal;

perform linear predictive coding (LPC) analysis on said input speech signal to obtain a z-domain representation of said input speech signal;

compute a spectrum tilt and a noise-to-signal ratio (NSR) of said z-domain representation of said input signal;

obtain a spectrum tilt of a background noise model;

apply a gain to reduce energy of said input speech signal when said NSR is high;

~~reducing the~~ reduce a spectral valley energy of said input speech signal when said spectrum tilt of said input speech signal is close or equivalent to said spectrum tilt of said background noise model; and

apply an inverse filter to said input speech signal when said spectrum tilt of said input speech signal is not ~~equivalent~~ close to said spectrum tilt of said background noise model, wherein said inverse filter is an inverse of said z-domain representation of said background noise model.

6. (Previously Presented) The computer program product of claim 5, wherein said input speech signal comprises a plurality of sub-frames processed in sequence.

7. (Currently Amended) The computer program product of claim 5, wherein said gain is adaptive~~ly~~ based on characteristics of said input speech.

8. (Previously Presented) The computer program product of claim 5, wherein said background noise model is a first order model.

9. (Currently Amended) An apparatus for suppressing <u>background</u> noise from a speech signal, said apparatus comprising:

an object for receiving an input speech signal;

an object for performing linear predictive coding (LPC) analysis on said input speech signal to obtain a z-domain representation of said input speech signal;

an object for computing <u>a</u> spectrum tilt and <u>a</u> noise-to-signal ratio (NSR) of said z-domain representation of said input signal;

an object for obtaining <u>a</u> spectrum tilt of a background noise model;

an object for applying a gain to reduce energy of said input speech signal when said NSR is high ;

<u>an object for</u> reducing ~~the~~ <u>a</u> spectral valley energy of said input speech signal when said spectrum tilt of said input speech signal is <u>close or</u> equivalent to said spectrum tilt of said background noise <u>model</u>; and

an object for applying an inverse filter to said input speech signal when said spectrum tilt

of said input speech signal is not ~~equivalent~~ close to said spectrum tilt of said background noise model, wherein said inverse filter is an inverse of the z-domain representation of said background noise model.

10. (Previously Presented) The apparatus of claim 9, wherein said input speech signal comprises a plurality of sub-frames processed in sequence.

11. (Currently Amended) The apparatus of claim 9, wherein said gain is adaptive~~ly~~ based on characteristics of said input speech.

12. (Previously Presented) The apparatus of claim 9, wherein said background noise model is a first order model.

13. (New) The method of claim 1, wherein applying said gain, reducing said spectral valley energy and applying said inverse filter are performed using $g \,.\, [1/Fn(z/a)] \,.\, Fs(z/b)/Fs(z/c)$, wherein parameters a ($0<=a<1$), b ($0<b<1$), and c ($0<c<1$) are adaptive coefficients, and parameter g is an adaptive gain.

14. (New) The method of claim 13, wherein said parameters a, b, c, and g are controlled by said NSR.

15. (New) The computer program product of claim 5, wherein said computer readable program code to apply said gain, reduce said spectral valley energy and apply said inverse filter

are performed using g . [1/Fn(z/a)] . Fs(z/b)/Fs(z/c), wherein parameters a (0<=a<1), b (0<b<1), and c (0<c<1) are adaptive coefficients, and parameter g is an adaptive gain.

16. (New) The computer program product of claim 15, wherein said parameters a, b, c, and g are controlled by said NSR.

17. (New) The apparatus of claim 9, wherein said objects for applying said gain, reducing said spectral valley energy and applying said inverse filter are performed using g . [1/Fn(z/a)] . Fs(z/b)/Fs(z/c), wherein parameters a (0<=a<1), b (0<b<1), and c (0<c<1) are adaptive coefficients, and parameter g is an adaptive gain.

18. (New) The apparatus of claim 17, wherein said parameters a, b, c, and g are controlled by said NSR.